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CLAIMS:

What is claimed is:

1. A head end controller that controls a plurality of service units in a telecommunications system with a multi-carrier transmission scheme, the head end controller comprising: a logic circuit that generates control messages for the plurality of service units, wherein each service unit is assigned to at least one subband of a transmission bandwidth and wherein each subband includes a number of payload channels and at least one control channel; and a control channel transceiver that is adapted to be coupled to a distribution network of the transmission system, wherein the control channel transceiver broadcasts control messages to the plurality of service units over the control channels in the plurality of subbands.

2. The head end controller of claim 1, wherein the control messages include a personal identification number that identifies at least one service unit.

3. The head end controller of claim 1, wherein the control transceiver is adapted to be coupled to a hybrid fiber/coax network.

4. The head end controller of claim 1, wherein the logic circuit is adapted to detect collisions on the control channels of the plurality of subbands and is further adapted to instruct the service units to retransmit upstream messages on the control channel when a collision is detected.

5. The head end controller of claim 1, and further comprising a modem, coupled between the control channel transceiver and the distribution network, that is adapted to transmit the control signals to the service units over the control channels.

various delivery approaches have been proposed, such as optical fiber links to every home, direct satellite transmission, and wideband coaxial cable. However, these approaches are often too costly, and cheaper alternatives have emerged, such as the cable modem which uses existing coaxial cable connections to homes and various high bit rate digital subscriber line (DSL) modems which use the existing twisted-pair of copper wires connecting a home to the telephone company central office (CO).

Brief Summary Text (6):
Severed digital subscriber lines (DSL) technologies have been developed for different applications. The original 2B1Q Digital Subscriber Line technology has been used as the ISDN Basic Rate Access channel U-interface. The High-bit-rate digital subscriber lines (HDSL) technology has been used as the repeatertess T1 service.

Brief Summary Text (7):
An example of prior art use of DSL techniques is the Asymmetrical Digital Subscriber Line (ADSL) signaling for the telephone loop that has been defined by standards bodies as a communication system specification that provides a low-rate data stream from the residence to the CO (upstream), and a high-rate data stream from the CO to the residence (downstream). The ADSL standard provides for operation without affecting conventional voice telephone communications, e.g. plain old telephone service (POTS). The ADSL upstream channel only provides simple control functions or low-rate data transfers. The high-rate downstream channel provides a much higher throughput. This asymmetrical information flow is desirable for applications such as video-on-demand (VOD).

Brief Summary Text (8):
ADSL modems are typically installed in pairs, with one of the modems installed in a home and the other in the telephone company's central office servicing that home. The pair of ADSL modems are connected to the opposite ends of the same twisted-par and each modem can only communicate with the modem at the other end of the twisted-pair; the central office will have a direct connection from its ADSL modem to the service provided (e.g., movies, Internet, etc.). FIG. 2a heuristically illustrates an ADSL modem (FIG. 2a uses "DSL" rather than "ADSL" for the modem) installed in the central office and one in the consumer's home, either a personal computer or a TV set-top box. Because an ADSL modem operates at frequencies higher than the voice-band frequencies, an ADSL modem may operate simultaneously with a voice-band modem or a telephone conversation.

Brief Summary Text (9):
A typical ADSL-based system includes a server located at the CO capable of providing movies or other data-intensive content, and a set-top-box at the residence that can receive and reassemble the data as well as send control information back to the CO. Meaning display or use of the downstream content typically requires a sustained data rate through the modem. Due to the sustained data rate requirements, ADSL systems are primarily designed to function under certain operating conditions and only at certain rates. If a subscriber line meets the quality requirements, the ADSL modem can

function, otherwise new line equipment must be installed, or line quality must be improved.

Brief Summary Text (11):
An ADSL modem differs in several respects from the voice-band modems currently being used for digital communication over the telephone system. A voice-band modem in a home essentially converts digital bits to modulated tones in the voice-band (30 Hz to 3.3 KHz), and thus the signals can be transmitted as though they were just ordinary speech signals generated in a telephone set. The voice-band modem in the receiving home then recovers the digital bits from the received signal. The current ITU V-series voice-band modem standards (e.g. V.32 and V.34) call for transmission at bit rates of up to 33.6 Kbps; even these rates are far too slow for real-time video and too slow for Internet graphics. In contrast, an ADSL modem operates in a frequency range that is higher than the voice-band; this permits higher data rates. However, the twisted-pair subscriber line has distortion and losses which increase with frequency and line length; thus the ADSL standard data rate is determined by a maximum achievable rate for a length of subscriber lines, e.g. 9,000 feet (9 ft) for 26 gauge lines, or 12 kft for 24 gauge lines.

Brief Summary Text (12):
Voice-band modem data speeds are limited by at least the following factors: 1) the sampling rate of the line cards in the central office is only 8 KHz; 2) the low bit resolution of the A/D and D/A converters used on the line cards reduces dynamic range; and 3) the length of the subscriber line (twisted-pair) and any associated electrical impairments. Although an ADSL modem avoids the first two factors, it also suffers from subscriber line length limitations and electrical impairments. FIG. 4c illustrates how the capacity of a subscriber line decreases with increasing line length for the two existing wire sizes. A similar capacity decrease with length applies to any type of twisted-pair subscriber line modem.

Brief Summary Text (13):
FIG. 4a shows in block format a simple ADSL modem whose transmit hardware 30 includes the bit encoder 36, inverse fast Fourier transform 38, P/S 40, digital-to-analog converter 42, filter and line driver 44 for transmission and transformer 46. The receive portion 32 includes a transformer and filter 48, analog-to-digital converter 50, an equalizer for line distortion compensation 52, S/P 54, fast Fourier transform 56, and bit decoder 58. An echo cancellation circuit from the transmission portion to the reception portion may be included to suppress signal leakage. The ADSL standard uses discrete multitone (DMT) with the DMT spectrum divided into 256 4-KHz carrier bands and a quadrature amplitude modulation (QAM) type of constellation is used to load a variable number of bits onto each carrier band independently of the other carrier bands.

Brief Summary Text (14):
The number of bits per carrier is determined during a training period when a test signal is transmitted through the subscriber line to the receiving modem. Based on the measured signal-to-noise ratio of the received signal, the receiving modem determines the

optimal bit allocation, placing more bits on the more robust carrier bands, and returns that information back to the transmitting modem.

Brief Summary Text (15):
The modulation of the coded bits is performed very efficiently by using a 512-point inverse fast Fourier transform to convert the frequency domain coded bits into a time domain signal which is put on the twisted-pair by a D/A converter using a sample rate of 2.048 Mhz (4.times.512K). The receiving ADSL modem samples the signal and recovers the coded bits with a fast Fourier transform.

Brief Summary Text (16):
Discrete multi-tone (DMT) has been chosen as the line code for the ADSL standard. A typical DMT system utilizes a transmitter inverse FFT and a receiver forward FFT. Ideally, the channel frequency distortion can be corrected by a frequency domain equalizer following the receiver FFT. However, the delay spread of the channel in the beginning of the receiver FFT block contains inter-symbol interference from the previous block. As this interference is independent of the current block of data, it can not be canceled just by the frequency domain equalizer. The typical solution adds a block of prefix data in front of the FFT data block on the transmitter side before the block of FFT data is sent to the D/A. The prefix data is the repeat copy of the last section of FFT data block.

Brief Summary Text (17):
On the receiver side, the received signal is windowed to eliminate the cyclic prefix data. If the length of the channel impulse response is shorter than the prefix length, inter-symbol interference from the previous FFT data block is completely eliminated. Frequency domain equalizer techniques are then applied to remove intra-symbol interface among DMT subchannels. However, since the channel impulse response varies on a case by case basis, there is no guarantee that the length of the impulse response is shorter than the prefix length. An adaptive time domain equalizer is typically required to shorten the length of the channel response within the prefix length.

Brief Summary Text (19):
The following patents are related to DMT modems: U.S. Pat. No. 5,400,322 relates to bit allocation in the multicarrier channels; U.S. Pat. No. 5,479,447 relates to bandwidth optimization; U.S. Pat. No. 5,317,596 relates to echo cancellation; and U.S. Pat. No. 5,285,474 relates to equalizers..

Brief Summary Text (20):
Alternative DSL modem proposals use line codes other than DMT, such as QAM, PAM, and carrierless AM/PM (CAP). Indeed, ISDN uses a 2bit-1quaternary (2B1Q) four level symbol amplitude modulation of a carrier of 160 KHz or higher to provide more data channels.

Brief Summary Text (21):
CAP line codes typically use in-phase and quadrature multilevel signals which are filtered by orthogonal passband filters and then converted to analog for transmission. FIG. 4b shows a block diagram

for the transmitter 321 and receiver 325 of a DSL modem using the CAP line code and including both an equalizer 750 and echo cancellation 327.

Brief Summary Text (22):
The following patents are related to CAP modems: U.S. Pat. No. 4,944,492 relates to multidimensional passband transmission; U.S. Pat. No. 4,682,358 relates to echo cancellation; and U.S. Pat. No. 5,052,000 relates to equalizers.

Brief Summary Text (23):
Modems using CAP or DMT, or other line codes, essentially have three hardware sections: (i) an analog front end to convert the analog signals on the subscriber line into digital signals and convert digital signals for transmission on the subscriber line into analog signals, (ii) digital signal processing circuitry to convert the digital signals into an information bitstream and optionally provide error correction, echo cancellation, and line equalization, and (iii) a host interface between the information bitstream and its source/destination.

Brief Summary Text (24):
However, these DSL modems have problems including: 1) higher bit rates for video that cause them to be complicated and expensive; 2) their bit rates are optimized for a fixed distance, making them inefficient for short subscriber loops and unusable for long subscriber loops; and 3) either DMT or CAP operates better for given different conditions (e.g. noise, etc.) that may or may not be present in a particular subscriber loop to which the DSL modem is connected.

Brief Summary Text (26):
An alternative wired system proposes utilizing copper infrastructure and high speed modems to transmit digital two way data. These systems can operate with several modulation schemes including Carrierless Amplitude/Phase (CAP), Discrete Multitone (DMT), DWMT and Subscriber Loop Carrier (SLC). Asymmetrical Digital Subscriber Loop (ADSL), Very-High-Data-Rate Digital Subscriber Line (VDSL) and High-Data-Rate Digital Subscriber Line (HDSL) modems currently under development will offer different data rates to carry communication signals to and from the customer premises. For copper wire based systems Limited bandwidth, signal attenuation resulting from the wire gauge and transmission distance all decrease such possible system data rates. Integration into the copper twisted pair network can be active or passive. To maintain the high data rates capabilities of these systems amplifiers will be required to maintain the signal strength and condition between communication points.

Brief Summary Text (30):
The present invention provides a new high speed modem for use on standard telephone twisted-pair lines at lengths of up to 21,000 ft. This new modem will be referred to as MDSL, mid-band digital subscriber line. The MDSL modem of the present invention makes use of frequency division multiplexing (FDM) to separate the downstream and upstream transmitted signals. Although the modulation scheme for MDSL can be arbitrary, two specific modulation schemes that may

be employed are QAM/CAP and Discrete Multitone (DMT). A startup procedure for achieving synchronization between the MDSL modem of the present invention at the central office (CO) and the MDSL modem at the remote user (RU) end is provided as part of the present invention.

Brief Summary Text (32):
The present invention provides a modem which supports both voice-band and above voice-band (DSL) functionality using preselected common circuitry. Preferred embodiments use a DSP to run either voice-band or above-voice-band modem software in combination with, either separate or combined analog front ends, and a common host interface (either serial or parallel). The same internal components may be employed for either the voice-band or the above-voice-band modem, and the modem may have an integral splitter to separate the voice-band for use by a telephone set.

Brief Summary Text (33):
The present invention provides a programmable Digital Signal Processor (DSP) implementation approach that allows different existing ADSL line codes, Discrete Multitone (DMT) and Carrierless AM/PM (CAP), to be implemented on the same hardware platform as a voice-band modem. With a DSP implementation, the desired transmission rate can also be negotiated in real time to accommodate line condition and service-cost requirements.

Brief Summary Text (34):
This line code and rate negotiation process can be implemented at the beginning of each communication session through the exchange of tones between modems at both ends. A four-step Mid-band Digital Subscriber Lines (MDSL) modem initialization process is used for line code and rate compatibility.

Brief Summary Text (35):
Although Digital Subscriber Line (DSL) signaling is used to convey digital data over existing twisted-pair copper telephone lines connecting the telephone company central office (CO) to residential subscribers, conventional DSL data modems are designed to provide service to a certain percentage of residential customers at a prescribed data rate. A new rate negotiation method of the present invention enables a variable-rate DSL (VRDSL) system. Using the rate negotiation method, the variable rate system adapts its throughput based on line conditions, computational capabilities, network accessibility, and application requirements. This service can be added to a telephone subscriber loop without disrupting the plain old telephone service (POTS). Hence, a voice-band modem connection can also be made available independent of the DSL connection.

Brief Summary Text (36):
The rate negotiation method provides systematic control for a DSL system that supports multiple rates. The data rates can be varied depending on modem cost, line conditions, or application requirements. The modem functions as a variable rate data link capable of supporting many different applications, including VOD, videophone, multiple ISDN links, and new network access applications. By considering the capability of a particular DSL

connection, available computational power, and any special application program requirements, the data rate can be adapted by the negotiation method to a suitable level. This scheme provides symmetrical or asymmetrical data links and supports simultaneous applications requiring arbitrary mixes of symmetrical and asymmetrical links. A part of the symmetrical portion of the DSL transmission throughput can be used for telephone calls or video telephone calls. A part of the asymmetrical portion of the DSL transmission throughput can be used for internet access or VOD services. The rate negotiation method supports many different network applications using DSL.

Brief Summary Text (37):
The typical implementations of DSL modems, thus far, have supported only connectionless services between the subscriber and the network. However, since DSL is terminated at the local central office, a telephone-network friendly DSL interface is desirable. To facilitate multiple virtual service connections, an operations/signaling channel, similar to the ISDN D channel, is preferred for exchanging service and control messages. A preprocessor in the CO-end DSL modem is also necessary to collect operational messages before passing signaling and data packets to the CO control-channel server.

Brief Summary Text (38):
The DSL modem of the present invention supports connectionless as well as connection-oriented (switched) services.

Brief Summary Text (39):
The method of rate negotiation is preferably employed with a DSL system capable of a varying rate. An example is a viable-rate DSL (VRDSL) system that can provide a variable upstream transmission throughput up to 400 Kbps and a downstream transmission throughput of from 400 Kbps up to 2.048 Mbps. (However, the invention is not constrained to vary within the rates given by this example system.) With lower throughput, operation with poor line conditions is supported. Lower data rates also allow the design of less expensive modems for less demanding applications. This is consistent with the mid-band DSL (MDSL) design philosophy of the present invention, which can provide a symmetrical 400 Kbps link using the same hardware platform as a voice-band modem. With high downstream throughput, VRDSL can be made compatible with ADSL. Basically, the VRDSL rate negotiation method provides the capability to serve a range of price/performance DSL modems that can m maximize throughput based on individual line conditions and processing power. In VRDSL signaling, the POTS will still be available through the same telephone subscriber loop.

Brief Summary Text (42):
The line connection management process for a mid-band digital subscriber lines (MDSL) provides a simple, efficient and flexible interface to manage the line connection between MDSL-C (MDSL in Central Office site) and MDSL-R (MDSL in residential site). MDSL uses four different line modes: leased line with single link (LLSL); leased line with multiple links (LLML); switched line with soft dial (SLSD); and switched line with hard dial (SLHD). The host interface for the LLSL mode, has three different line states: line

drop, line disconnected and line connected. An internal state machine of the MDSL modem can record and monitor the line status and notify the state change to the other MDSL modem, as well as the host processor. The protocol used for exchanging line connection management messages of the present invention is a simplified point-to-point link control protocol.

Brief Summary Text (45):
The present invention also provides a simple algorithm to train the time domain equalizer of an MDSL modem. By the same procedure, the FFT frame boundary is also reliably detected.

Brief Summary Text (46):
This invention also provides point-to-multipoint delivery of communication services and more particularly to distribution methods which integrate wire and wireless systems via modems into an efficient digital signal distribution network designated a Hybrid Wireless Wire-Line Network (HWWN). A key element included in this system architecture is the bandwidth management feature which provides for efficient use of the available spectrum based on user demands for data rates and channel transmission conditions.

Brief Summary Text (47):
The invention also provides a direct equalizer system with an adaptive filter in the transmitter for symmetrical dispersive transmission channels. The direct equalization approach avoids the use of an expensive high precision high sampling rate A/D converter and a high precision adaptive filter in the receiver. In the transmitting data path the adaptive filter only needs a precision equal to the symbol bit resolution. The filter coefficients are identified in the receiving path using a sign LMS algorithm (which only involves shift and addition operations). Thus, the direct equalizer system of the present invention is an inexpensive approach for the realization of high data rate transmission systems over symmetrical dispersive channels.

Drawing Description Text (3):
FIGS. 1a-e show a preferred embodiment multimode modem.

Drawing Description Text (4):
FIGS. 2a-h show preferred embodiment modem Central Office modems and distribution systems;

Drawing Description Text (5):
FIGS 3a-e show preferred embodiment modem applications and ISDN signaling;

Drawing Description Text (6):
FIGS. 4a-c show prior art modems plus subscriber line capacity;

Drawing Description Text (7):
FIGS. 5a-b show another preferred embodiment modem;

Drawing Description Text (13):
FIGS. 11a-n show preferred embodiment modem driver;

Drawing Description Text (16):

FIGS. 14a-e show preferred embodiment modem pool;

Drawing Description Text (17):
FIG. 15a shows a channel transfer function of a 24 gauge 50 meter twisted pair;

Drawing Description Text (18):
FIG. 15b shows an eye pattern without channel distortion compensation;

Detailed Description Text (2):
Overview of preferred embodiment modems

Detailed Description Text (3):
FIG. 1a shows a functional block diagram of a first preferred embodiment of a multimode modem 100 of the present invention. In FIG. 1a, modem 100 includes both a voice-band and DSL band data path to a single subscriber line (twisted-pair) 140, which connects to a telephone company central office. A voice-band analog front end (VB AFE) 110 transmits and receives at frequencies in the voice-band (30 Hz to 3.3 Khz), whereas the digital subscriber line analog front end (DSL AFE) 120 transmits and receives at frequencies above the voice-band (above 4 KHz). A Splitter 130 connects to the subscriber line 140 and separates the incoming signals into its voice-band and above-voice-band components. POTS (plain old telephone service) occurs in the voice-band and a telephone may be connected to the subscriber line directly or through the splitter 130.

Detailed Description Text (4):
Modem 100 utilizes a single programmable digital signal processor (DSP) 150 as part of the DSL band data path and as part of the voice-band data path, but typically uses two separate data input ports. Generally, the DSL band will have a much higher bit rate than the voice-band data path, so using separate DSP ports will be more convenient than using a single port with a buffered multiplexer; although the use of such a multiplexer is an alternative clearly within the scope of the present invention. For example, the DSL band operation modem 100 may employ an upstream (from residence to central office) frequency band centered at 100 KHz with a total bandwidth of slightly less than 200 KHz, and a downstream (from central office to residence) frequency band centered at 300 KHz and also of total bandwidth slightly less than 200 KHz; this frequency allocation provides for full duplex operation of modem 100. Generally multiple DSPs, instead of a single DSP, may be employed to increase functions performed or to increase performance. The DSP 150 is connected to a host interface circuit 160.

Detailed Description Text (5):
Modem 100 can select from multiple line codes and, further, modem 100 can perform as either a high-bit-rate DSL modem in frequencies above voice-band or as a voice-band modem (such as V.34), either simultaneously or consecutively just by switching programs being executed by the DSP 150. The various line code programs can be stored in the DSP onboard memory or in auxiliary memory not shown in FIG. 1a. Also, alternative line codes for the DSL modem

operations (e.g., a CAP or a DMT line code) can be used, again depending upon the program executed by the DSP 150.

Detailed Description Text (6):
FIGS. 1b-c illustrate the DSL data path portion of modem 100 which includes analog-to-digital 172 and digital-to-analog 170 converters, filters 174, 176, a transmission driver 178, and a receiver amplifier 180. FIG. 1b additionally explicitly shows a phase locked loop 182 clock generator that synchronizes the modems' internal clocks with the clock signals from the host (or the central office). FIG. 1c omits the bandpass filters and instead shows various optional memory types, both SRAM 184 and nonvolatile EEPROM 186 which could hold line code programs. When modem 100 acts as a voice-band modem, the spitter 130 provides the voice-band frequencies to the voice-band analog front end 120.

Detailed Description Text (7):
FIG. 1d illustrates the DSP software for modem 100 in DSL mode and includes (i) an optional kernel (operating system) 190 for the DSP, (ii) host interface 192, (iii) optional management maintenance control 194, (iv) framing 196, (v) embedded operations control 198, (vi) channel multiplexer 199 for multiplexing the embedded operations control with the data stream, (vii) scrambler logic 191 for bitstream scrambling (viii) the transceiver logic 193 such as a CAP or DMT logic which includes the bits-to-symbols conversions, equalization, echo cancellation, and (ix) modulator/demodulator 195 logic and optional forward error correction (FEC).

Detailed Description Text (8):
FIG. 1e illustrates the software protocol hierarchy for applications running on modem 100 interfacing with a host. The physical layer 185 (layer 1) includes the DSP software for modulation, bitstream scrambling, and multiplexing control signals with the data stream. The data link layer 187 (layer 2) in the DSP includes embedded operations control and framing. The network layer 189 (layer 3) in the host includes the modem driver (e.g. NDIS type for a Windows 95/NT) and transport protocols such as PPP (point-to-point protocol). Applications such as internet browsers interact with the transport protocols.

Detailed Description Text (9):
For voice-band modes of operation, modem 100 may use software similar to standard voice-band modems (e.g. V.34, etc.).

Detailed Description Text (10):
The present invention provides a new high speed modem 100 for use on standard telephone twisted-pear lines at lengths up to 21,000 ft. This new modem 100 will be referred to as MDSL, mid-band digital subscriber line. The MDSL modem 100 makes use of frequency division multiplexing (FDM) to separate the downstream and upstream transmitted signals. Although the modulation scheme for MDSL can be arbitrary, two specific modulation schemes that may be employed are QAM/CAP and Discrete Multitone (DMT). A startup procedure for achieving synchronization between the modem at the central office (CO) and the modem at the remote user (RU) end is provided as part of the invention.

Detailed Description Text (11):
One of the modulation schemes selected for one embodiment of the MDSL modem is Carrierless AM/PM (CAP). CAP can be considered as a special case of the more conventional Quadrature Amplitude Modulation (QAM). The main difference is that CAP performs most of its processing in the passband, while QAM performs most of its processing at baseband.

Detailed Description Text (15):
FIG. 2a shows modem 100 in a home 210 communicating with another modem 100 in the central office 220. This central office 220 modem 100 may have various capabilities and loads, and the subscriber loop 140 may be in a particular condition, so the modems execute an initialization process to select the line code (CAP, DMT or others), the bit-rate, and train the equalizers. Then the modems begin data communication.

Detailed Description Text (16):
FIGS. 2b-c illustrate alternative central office connections to subscriber lines with DSL modems: each subscriber line has a DSL AFE (analog front end) and an analog switch connects an AFE output to a DSL processor, either a DSP similar to the DSP in the residence modem or a single DSP for multiple AFEs. The central office monitors the AFE outputs and a digital switch assigns an available DSP to communicate with the corresponding residence DSL modem. The central office polls the AFEs to find active modems in the residences. As FIGS. 2b-c show, the central office DSL modem connects to a remote access server on a local area network with packetized information (e.g., Internet) or a wide area network with constant bit rate data which is sent directly across the public switched telephone network trunk lines. The information sent by the residence modem would be identified or signaled via an out of band signaling method (e.g. similar to ISDN Q.931 signaling), rather than an off-hook signal, plus telephone number sent in the voice-band to the analog switching and line cards. FIG. 2c illustrates the major functional blocks of a central office DSL modem (the DSL band is already separated from the voice-band) as an AFE 240, DSP 260, Communications Controller 280 and ARM or RISC processor 290. The modem has a connection to both the constant bit rate transmissions (voice, video conferencing, etc.) being forwarded to a time division multiplexed (TDM) bus and packetized data (Internet, Intranet, private networks, etc.) being forwarded to a control bus (and then to the trunk lines). FIG. 2c depicts the terminology "xDSL" which may be ADSL or any other type of DSL modem. These various functions could be all performed in a single DSP 260.

Detailed Description Text (18):
Referring now to FIG. 2d there may be seen a simplified functional block diagram of an architecture of the present invention for a hybrid wireless wire-line network (HWWN) 2000. More particularly, an architecture and a method that distributes telephony, television and data signals via an integrated transmission network is depicted in FIG. 2d. Communication distribution begins at the headend 2002 or central office 2004. Signals are digitized and may be sent via an optical feeder link 2006 to a wireless distribution node 2008. Various techniques can be employed to modulate the RF carrier which

is upconverted for transmission to the neighborhood. Remote terminals called Wireless Network Units (WNU) 2010 may be deployed in the neighborhood and use antennas to receive the Radio Frequency (RF) signals, translate them to Intermediate Frequencies (IF) then to a low carrier frequency signal coupled onto a Digital Subscriber Line (DSL) and transported via a Very-High-Data-Rate Digital Subscriber Line (VDSL) or MDSL on the twisted pair 2012 to a residence 2014. In a two-way system these antennas will be part of the return connection platform to transmit information back from the customer premises 2014 to the node 2008. Twisted copper pair lines or coaxial cables via high speed modems transmit or receive the digital signals initializing or completing the transmission network at the customer premises. Network control and routing functions are accomplished via an appropriate control channel. The present invention uniquely utilizes the capabilities of high speed modems and established wired and wireless distribution technology in an integrated transmission network. Additionally, bandwidth can be dynamically controlled and frequencies reused to optimize the transmission network. Based on user demands and detected interference the system management adjusts the data rates to optimize network performance. System management is achieved by passing information through the Operation Support System (OSS).

Detailed Description Text (19):
In accordance with the preferred embodiment Hybrid Wireless Wire-Line Network (HWWN), a method of broadband communication distribution combines the advantages of wireless distribution while integrating the digital signals back into the existing copper or coaxial network at a Wireless Network Unit (WNU) 2010. The final transmission link to the customer premises is made using a VDSL (or MDSL) line driver to the VDSL (or MDSL) receiver. System management is employed to dynamically adjust bandwidth based on customer data rate requirements. Information selection and channel quality are monitored and controlled via the control channel and Operation Support System (OSS). Various architectures link the network data communications systems together through the seven Open System Interconnect functional layers.

Detailed Description Text (20):
The HWWN method of distribution affords cost and performance advantages and eliminates many of the disadvantages of the other systems mentioned above. Specifically, by using a wireless point to multipoint system combined with modems, higher data rates can be provided over longer distances with reduced bit error rate (BER). Additionally, the wireless feature allows for a rapid deployment with increased capacity added on as required. Modems provide access to multiple customers from a wireless network unit. This integrated architecture increases customer access over systems offering direct distribution to the customer premises. Using this architecture a single wireless network unit can provide an interface to connect to several hundred customer premises. The network architecture of the present invention enables such features as higher speed World Wide Web access, video conferencing and supports 10 Base T Ethernet, 100 Base T Ethernet and Asynchronous Transfer Mode (ATM) connection to the customer premises at an effective cost.

Detailed Description Text (21):

Various architecture embodiments may be deployed using a variety of modulation techniques. For illustrative purposes, a higher level modulation scheme, such as 64 QAM will be utilized to make effective use of any available spectrum. In wireless systems degradation in the signal to noise resulting from things like multipath, and adjacent channel carrier can cause signal interference. Adaptive equalization can correct for some of these problems. Sectorized antennas at the transmitting node with alternating frequency and alternating antenna polarization can offer increased channel densities with reduced signal interference. To reduce interference caused by the return path, Quadrature Phase Shift Keying (QPSK) modulation may be incorporated with adaptive channel band control and spatial diversity to reduce system interference.

Detailed Description Text (23):
FIG. 2d is a block diagram of a presently preferred network embodiment comprised of a wireless point to multipoint system coupled into a conventional copper telephony system. Another network embodiment might employ a bus architecture for deployment into a coaxial system or with a satellite feeder as the node. The wireless system is made up of multiple nodes such as node 2008 in FIG. 2d. Enough Wireless Network Units 2010 are deployed to cover the desired service area. Terrestrial network deployment and integration depends on the location of the central office, headends, and access to node sites, buildings or towers. However, any actual configuration depends on the number of customers and the required data rates. At the central office, modems feed a concentrator and packetizer into the appropriate data stream. Multiple modems multiplexed at the central office send data stream via fiber optical terminal (FOT) over an optical link to a remote node site for transmission over the wireless node antenna. Similarly, the video headend integrates the video streams onto a FOT which links to the node for transmission over the wireless node antenna. WNU equipment receives the transmission and translates the signal down for distribution to the end customer.

Detailed Description Text (24):
To establish effective communication using a higher level of modulation, such as for example but not limited to a 64 QAM modulation scheme, several techniques can be utilized to decrease the effect of interference. Referring to FIG. 2e the node 2008 antenna can be deployed to cover a complete 360.degree. cell in a sectorized pattern. FIG. 2e shows 4 nodes 2008a-2008d, with a transmitting tower at the center of each circle. With each node tower or platform, antennas are arranged in sectors. For the purpose of this discussion sectors are shown as 60.degree. sectors. This sectorized pattern is then repeated around the node and in the adjacent cells. These sectors may be deployed with alternating horizontal and vertical polarization and the communication area can provide coverage with significantly less interference. To further reduce the interference, transmit frequencies can be alternated from sectors. The disadvantage of this method is it decreases the number of channels available for transmission of information to the customers. The 60.degree. sectors counters this effect by providing for a high level of frequency reuse and thus boosting the channel capacity.

Detailed Description Text (25):
Table 1 shows channel capacity vs. modulation type and the effect of sectorizing. For illustrative purposes, a 3 Mbps transmission channel was selected. As can be seen in the table, higher levels of modulation such as 64 QAM with forward error correction coding, Reed-Solomon outer code for burst error protection and Trellis inner code at the symbol level, provide a higher bandwidth efficiency. The table shows the number of 3 Mbps channels that each modulation technique can support given a total bandwidth of 780 MHz. To reduce the interference and meet the higher system signal to noise ratio required for 64 QAM modulation, channel frequencies would most likely have to alternated from sector to sector. This would not be the case for a QPSK system due to the lower signal to noise requirements. Taking into account the alternating frequency plan for higher efficiency modulation schemes Table 1 shows the practical number of channels which can be obtained and concludes with the effect on channel capacity of deploying 6 sectors per node. Various other system factors including linearity, signal to noise ratio, effective isotropic radiated power (EIRP), and phase stability coupled with receiver noise figure, antenna size, system gain with adequate path link margin will determine which technique provides the most cost effective system.

Detailed Description Text (26):
FIG. 2f shows a block diagram of a WNU 2010 and the end customer modem equipment. Downstream RF channels carrying multiplexed subcarrier signals are selected and received at the antenna, converted to IF, demodulated and demultiplexed. Using a VDSL (or MDSL) line driver data is coupled via a splitter for separation of the voice and DSL signals. Data is sent via a low carrier frequency, Quadrature Amplitude modulated (QAM) signal over the twisted pair line. To complete the downstream path the VDSL modem receives the digital signals and translates the signal back into information.

Detailed Description Text (27):
FIG. 2f also shows the upstream return path from the customer premises 2014 to the WNU 2010. The digital signals are sent upstream via the VDSL transmitter over the twisted copper pair and are received by the VDSL receiver located in the WNU. Digital upstream channels are multiplexed, encoded and converted to RF frequencies for transmission to the Node receiver.

Detailed Description Text (28):
FIG. 2g details the WNU 2010 operational blocks. The data coupled onto the existing copper line are transmitted via the Very-High-Data-Rate Digital Subscriber Line (VDSL) at baseband to and from the customer premises. The control channel has three primary functions 1) pass channel selection information, 2) allocate bandwidth and 3) analyze channel interference resulting in bit error rate. As part of the first function, tuners are located in the WNU to tune to the appropriate channel. Broadcast information can be shared via multiple VDSLs. This acts as a virtual tuner reduces equipment costs. For the second channel control function, bandwidth allocation, data rate requests are sent via control signals to the WNU from the customer premises modem.

The WNU forwards the request to the Node where capacity allocation is arbitrated and assigned. If insufficient system resources are available the system will negotiate other user rates in an attempt to complete the newly requested link. This information is managed at the network management layer and can be used to bill customers based on actual data rate used. Communications originating at the node utilize the management layer to determine the customer selected data rate and based on the communication segment requirements the node would only transmit on the channels required for the data rate. A fully populated node (all carrier frequency) could be realized using frequency diversity on the WNU and transmit node and spatial diversity at the WNU allowing for dynamic transmit and receive frequency allocation. This dynamic bandwidth allocation could be achieved through the use of variable or switched bandwidth filters thus reducing or eliminating the need for a guard band. Finally, function three analyzes the channel interference at any given time and improves the carrier to interference (C/I) by reducing the bandwidth. The effects of these last two techniques are to provide a system with a variable data rate capability resulting in a more efficient utilization of the spectrum.

Detailed Description Text (29):
The node receiver downconverts, demodulates, demultiplexes and interfaces the signals back into the switched telephone network for distribution. The control channel information is used to establish and prioritize communication link paths based on the type of information, arbitrate data rates, manages transmit and receive frequency separation and integrate the wireless into the OSS.

Detailed Description Text (31):
MMDS is a one way terrestrial video system. A HWWN could provide acquisition improvements similar to the satellite example. Again this embodiment could add two way high speed data capabilities and a second telephony line. Transmission of symmetrical payloads such as telephony require equal channel capacity in the transmit and receive modes. With the dynamic BER monitor and arbitrated data rates capability and digital compression techniques a HWWN system could be deployed which achieves two times capacity increases, or more. Some channel capacity could be used to support new applications such as high speed Internet connections. Additionally, the QAM modulation technique being considered for digital video MMDS systems could utilize sectorized nodes and manage channel allocation to reduce interference.

Detailed Description Text (32):
As a final embodiment, with a HWWN digital transmission architecture it is possible to develop a system to control and allocated the system bandwidth based on varying data capacity demands, type of information (data rates) and interference encountered. FIG. 2h summarizes such a system's capabilities. Assuming an 850 MHz frequency spectrum allocation, a QPSK modulation scheme with no concentration could provide 576 DSOs per 40 MHz RF channel. The data rate per 40 MHz channel is 37.056 Mbps accounting for overhead and pilot tones. Faster digital modems or sectorizing would increase these channel rates. A dynamically controlled HWWN system increases these rates by providing additional RF channel capacity. Based on utilization of the current

spectrum typically allocated for guard band dynamic channel allocation could provide an additional 3 RF channels. A HWWN digital transmission embodiment employing QAM modulation and interference measurement and control capabilities could potentially provide several more RF channels to increase capacity or provide higher data rates.

Detailed Description Text (34):
An alternative is for the central office to monitor each subscriber line with a DSL modem in the above-voice-band frequencies and when the line becomes active, an analog switch connects the subscriber line to a DSL modem in the central office. This mimics FIG. 2b except a simpler monitoring and an analog switch replace AFE monitoring and a digital switch. The same approach may also be used in conjunction with the local pedestal to shorten the subscriber line distance from residence DSL modem to the AFE on the central office end (physically located in the remote pedestal).

Detailed Description Text (35):
FIG. 3a shows a system with modem 100 in a personal computer 310 running Windows 95 (or Windows NT) with standard protocol stacks communicating over a subscriber line 140 with a corresponding modem 100 in the central office 220, which may be connected to an Internet access server via an Ethernet (10/100 Base T) interface. Modem 100 allows for both POTS or voice-band modem communication with another voice-band modem at the same time as the DSL portion of modem 100 connects to the Internet over the DSL portion.

Detailed Description Text (36):
Similarly, FIG. 3b shows a DSL modem acting as a router 330 for a local area network (LAN) 320 and coupling to devices 340, 342, 344 with corresponding DSL modems.

Detailed Description Text (37):
FIG. 3c shows half of a teleconferencing system based on modem 100 in a PC 350. Each teleconferencing end has modem 100 communicating at 384+16 Kbps with a modem in a central office 220. The central office modem transmits data between a concentrator and packetizer 360, and the packetizer converts to the 16 Kbps signaling channel into ISDN like signaling messages and applies the 384 Kbps stream to the T1/T3 service across the public switched telephone network. The central office 220 for the receiving party inverts these operations to feed the receiving modem 100. Traffic in the opposite directions proceeds similarly. Note that POTS can simultaneously be used with modems 100 for the voice in the teleconferencing. An analog delay can be inserted in the POTS output to synchronize with the video.

Detailed Description Text (38):
FIGS. 3d and 3e show ISDN-type signaling protocols and messages; modem 100 sends voice or data over the public switched telephone network. The SS7 network provides the backbone for carrying the ISDN user's part (ISUP) messages for call set-up and tear-down through the network.

Detailed Description Text (39):
FIG. 5a shows multimode modem 500, which includes the modem 100

features of both a DSL AFE 110 and a VB AFE 120, with a splitter 130 for subscriber line 140 connection together an ISDN front end 510 for connection to an ISDN line 142 plus an audio front end 520 for driving a speaker 146 and receiving a microphone 144 output as could be used for supporting a hands-free speakerphone. External RAM 530 may be nonvolatile (EEPROM or Flash EPROM) and/or volatile (SRAM or DRAM). The external RAM 530 may contain various programs for different line codes that may be used by the DSP 150. Such line codes may be DMT, QAM, CAP, PSK, FM, AM, PAM, DWMT, etc.

Detailed Description Text (40):
The transmit part of modem 100 consists of in-phase and quadrature passband digital shaping filters implemented as a portion of QAM transceiver logic; and the receive part consists of a fractionally spaced complex decision feedback equalizer (DFE) with in-phase and quadrature feedforward filters and cross-coupled feedback filters implemented as a portion of QAM transceiver logic. Optionally, the QAM transceiver logic may include a Viterbi decoder.

Detailed Description Text (41):
When modem 500 is active, modem 500 may provide voice-band modem functionality, DSL band modem functionality, ISDN functionality, audio functionality, other line code functionality, etc., or any combinations of the foregoing.

Detailed Description Text (42):
The present invention also includes a system where multiple like and different modems are simultaneously implemented in a single DSP hardware device. For example, voice-band (e.g., V.34), DSL, cable, terrestrial and other wireless and/or satellite modems are implemented simultaneously by the same DSP device. This is now becoming possible with increased processing capabilities of DSP devices. The advantages of this approach are to reduce overall system cost where the system requires multiple modems (e.g., Remote Access Systems (RAS): processing requirements are reduced due to reductions in processing overhead and program and data memory are reduced by sharing program and data memory buffers. For example, program memory is reduced when multiple like modems are executed simultaneously by a single DSP device. Interface and other miscellaneous glue logic are reduced by sharing the same logic between multiple modems, as well as better facilitating for statistical multiplexing and rate control.

Detailed Description Text (43):
In the near-term, the following situations win predominate, but these combinations will expand as DSP MIPS capabilities increase as a natural progression in the semiconductor industry: multiple voice-band modems in same DSP; voice-band and DSL modems in same DSP; voice-band and cable modems in same DSP; multiple DSL modems in same DSP; multiple cable modems in the same DSP; and/or any combination of the above.

Detailed Description Text (45):
Referring now to FIG. 6a, there may be seen a schematic diagram of the interconnection of a telephone 212 and modem 500 to a central office 220, via a subscriber loop 140.

Detailed Description Text (46):
Systems based on the DSL technology and available today are ISDN Basic Rate Access Channel and Repeaterless T1. DSL systems under development are Asymmetrical Digital Subscriber Lines (ADSL), Symmetrical Digital Subscriber Lines (SDSL), and Very-high-bit-rate Digital Subscriber Lines (VDSL). The transmission throughput of a DSL system is dependent on the loop loss, the noise environment, and the transceiver technology.

Detailed Description Text (49):
The transmission throughput of DSL for ISDN Basic Rate Access Channel is 160 Kbps. The transmission throughput of HDSL for repeaterless T1 is 800 Kbps. The transmission throughputs of ADSL are between 16 Kbps Ebps to 640 Kbps in the upstream (from a subscriber to a telephone central office) and between 1.544 Mbps to 6.7 Mbps in the downstream. The transmission throughputs of MDSL are presently believed to be 384 Kbps in the upstream and between 384 Kbps to 2.048 Mbps in the downstream.

Detailed Description Text (50):
A passband DSL system can be implemented with a single carrier using Quadrature Amplitude Modulation (QAM) or Carrierless AM/PM (CAP) line codes. A single carrier system depends on the adaptive channel equalizer to compensate for the channel distortion. The channel equalizer usually operates at a multiple of the signaling baud rate. FIG. 6c depicts a block diagram of a CAP transceiver.

Detailed Description Text (51):
More particularly, D/A 614 is connected to transmitter fitters 610, 612 and to filter 616. Filter 616 is connected to channel 620. Channel 620 is connected to filter 630 which is connected to A/D 632. A/D 632 is connected to equalizers 634, 636. A portion of the circuitry 638 recovers the time.

Detailed Description Text (52):
A DSL system can also be implemented with multiple carriers using the Discrete MultiTone (DMT) line code. A DMT system divides the channel into many subchannel carriers to better exploit the channel capacity and to reduce the channel distortion in addition to allowing for a relatively simple adaptive channel equalizer which only compresses the time spread of the channel impulse response rather than correcting it. A simple frequency domain equalizer completes the channel equalization. The signaling band rate of the DMT subchannels is much lower than the band rate of a single carrier system.

Detailed Description Text (53):
FIG. 6d depicts a block diagram of a DMT transceiver. More particularly, IFFT block 640 is connected to D/A 644, which is connected to transmit filter 646 which is connected to channel 650. Channel 650 is connected to alter 660 which is connected to A/D 632 which is connected to equalizer 664, which is connected to FFT block 666. Startup 642 and time recovery 668 circuitry is also included.

Detailed Description Text (54):
One MDSL modem embodiment uses frequency division fu duplex for

lower hardware cost and lower crosstalk noise level. Such an MDSL modem will provide a minimum of 384 Kbps full duplex transmission link between a central office and a subscriber for a loop length of up to 21 kft. Under favorable subscriber loop conditions, this MDSL modem can provide a much higher transmission throughput which is limited by channel capacity or the hardware capabilities of the subscriber-end modem. A full feature version of a subscriber-end MDSL modem communicates with ADSL modems at the central office end. The transmitter and receiver parts of the MDSL modem are capable of implementing either CAP or DMT line codes.

Detailed Description Text (55):
FIG. 6e depicts a block diagram of an MDSL modem 600. Modem 600 has a transmitter 676 connected to a D/A 674 which is connected to a filter 672 which is connected to hybrid circuit 670 which is connected to splitter 130. Hybrid circuit is also connected to filter 678 which is connected to A/D 680. A/D 680 is connected to receiver 682 which outputs the received signal. Timing recovery block 684 is used to recover the central office clock timing.

Detailed Description Text (56):
The purpose of the initialization process is to confirm the MDSL capability of the telephone subscriber loop 140 at both the central office 220 and the subscriber-end 210. The initialization process probes the channel 620, and produces information useful for transceiver training. The process then selects the line code, assuming multiple choices are available, and negotiates the transmission throughput based on the channel limit, traffic condition, or usage tariff.

Detailed Description Text (57):
The initialization process which is described later herein is: channel probing, line code selection, rate negotiation, and transceiver training.

Detailed Description Text (58):
An MDSL modem at the subscriber-end sends probing tones in the upstream band for a certain duration, with or without phase alternation for a part of these tones, according to a predefined time sequence. After the first time duration, the MDSL modem at the central office end responds with channel probing tones in the downstream band, again, with or without phase alternation for a part of these tones. This initial channel probing period may be repeated, if desired or necessary.

Detailed Description Text (59):
After the initial channel probing period, the MDSL modem at the subscriber-end has determined the line code capability of the central office end modem and has a channel model for the downstream band and, similarly, the MDSL modem at the central office end has determined the line code capability of the subscriber-end modem and has a channel model for the upstream band.

Detailed Description Text (60):
After the channel probing period, the MDSL modem at the subscriber-end should indicate/confirm its line code capability/preference by sending signature tones for a predefined

time duration. Similarly, the MDSL modem at the central office end should respond/confirm the line code selection by sending signature tones for a predefined time duration. This signature tone exchange process is preferably repeated for a limited number of times to determine a particular line code choice.

Detailed Description Text (61):
Another set of signature tones is then exchanged between MDSL modems at both ends for the transmission rate negotiation. The MDSL modem at the subscriber-end sends its rate capabilities and its preference. The MDSL modem at the central office end responds with its capabilities and its rate selection. MDSL modems determine a rate choice with a predefined rate change procedure described later herein. The transmission rate preference at the subscriber-end depends on the line condition, hardware capability, and user choice or application requirements. The transmission rate preference at the central office end depends on the line condition and the traffic load. Preferably, rate change during a communication session due to line condition change or user choice is allowed.

Detailed Description Text (62):
After the rate negotiation, the MDSL modems at both ends start transceiver training according to the conventional methods. Different time domain training sequences may be used for different line codes. It is an option to use the channel models obtained during the channel probing step to speed up the transceiver training process.

Detailed Description Text (64):
For simplicity, all frequency tones are assumed to be equally spaced with frequencies i.DELTA.f, amplitude a.sub.i, and phase .PHI..sub.i (usually it is either 0 or .pi.). At the receiver, the amplitude and phase of the received tones may be detected. The detected amplitude and phase of i-th frequency tone are b.sub.i and .phi..sub.i respectively. Assuming that there are N probing tones, the frequency response of the equivalent channel including filters at frequency i.DELTA.f is ##EQU1##

Detailed Description Text (65):
The impulse response of the equivalent channel can be calculated by a fast Fourier transform as ##EQU2## where T is the sampling period. The frequency spacing .DELTA.f depends on the spread of the channel impulse response. For a channel impulse response spread of n sampling periods,

Detailed Description Text (66):
where B is the total bandwidth of interest.

Detailed Description Text (67):
To distinguish from two different line codes, the phase of adjacent tones may be reversed by 180.degree. for one of the line codes. This line code could be DMT. To identify different line codes after channel distortion, select ##EQU3##

Detailed Description Text (68):
For a channel spread of 30 samples and a bandwidth of 100 Khz, select .DELTA.f.apprxeq.1.7 MHz and N as 64.

Detailed Description Text (69):
The channel probing tones should at least last more than a few times of the channel spread. With possible phase alternation, the channel probing tone duration should be 4 to 10 times of that necessary for the channel model recovery.

Detailed Description Text (84):
Tones can be generated by an IFFT operation as used for the DMT line code. A unit magnitude and zero/180.degree. phase vector signal is fed into the IFFT operation for the channel probing purpose. Selected zero phase vectors are used for the generation of signature tones.

Detailed Description Text (85):
Tones can be recovered by an FFT operation also as used for the DMT line code. The amplitude and phase information of each tone is recovered as a complex vector. A common phase difference due to the random sampling phase is calculated. Compensation produces a complex vector which is then used for calculating the channel transmission throughput and the channel impulse response, which might be used for transceiver training.

Detailed Description Text (86):
If the MDSL service is available through the telephone loop, the MDSL modem at the central office end should be on and monitor the upstream frequency band for probing tones.

Detailed Description Text (87):
Once power is on or a user service request is made, the MDSL modem at the subscriber-end sends upstream probing tones for a predefined time period and then monitors downstream probing tones. The MDSL modem at the central office end detects the probing tones, compensates for the random phase, stores them, and calculates the upstream channel transmission throughput. Meanwhile, the central office end MDSL modem sends the probing tones in the downstream frequency band.

Detailed Description Text (88):
The MDSL modem at the subscriber-end detects the probing tones, compensates for the random phase, stores them, and calculates the downstream channel transmission throughput. The subscriber-end MDSL modem then sends signature tones in the upstream band to indicate line code and transmission rate preferences.

Detailed Description Text (89):
The MDSL modem at the central office end detects the signature tones and responds with signature tones corresponding to its preferred offering. The subscriber-end MDSL modem then sends signature tones to confirm the offering or to request offering modification. The MDSL modems go into a transceiver training period after the confirmation of modem offering.

Detailed Description Text (90):
The throughput capacity of the DSL communication channel will change with line conditions and/or network accessibility. Line conditions dictate the achievable throughput of the physical

connection between the CO and the residence. Network accessibility describes the capability of the service provider's connection lining the DSL channel to the backbone network. The invented rate negotiation method incorporates a detailed understanding of the capacity-limiting factors of a DSL system.

Detailed Description Text (91):
DSL systems are traditionally engineered for the worst-case line condition for which service is to be provided. This approach simplifies the general installation procedure for telephone companies. However, restricting the DSL transmission throughput to that achieved in the worst-case line condition leaves most DSL systems operating well below their potential. The invented method provides a systematic procedure for maximizing the physical transmission throughput of each Local loop, enabling most DSL modems to operate at much higher rates than traditionally engineered. In fact, this method enables a majority of DSL modems to achieve a transmission throughputs which are only limited by the capabilities of the modem hardware. The rate negotiation method also provides time-varying adaptation in order to maintain the highest possible throughput as Line conditions or network accessibility changes.

Detailed Description Text (92):
The physical throughput of the twisted-pair DSL channel is limited by the receiver's ability to reliably distinguish the transmitted signal in the presence of noise and interference. The maximum possible throughput is upper bounded by the theoretical channel capacity of the physical link, such as depicted in FIG. 4c. The channel capacity of the link is determined by the bandwidth used, the received signal characteristics, and the noise and interference. The rate negotiation method will increase the DSL reach by providing low-rate options that can be supported by extremely long telephone subscriber loops while providing high-rate options that allow DSL modems operating on shorter loops to achieve a higher throughput.

Detailed Description Text (93):
The rate negotiation method considers the dynamic nature of the DSL transmission medium. The DSL is a time varying channel whose capacity may change due to improving/degrading channel conditions. As the channel conditions change, the theoretical maximum throughput also changes. The time-varying nature of the channel characteristics dictates the need for rate negotiation techniques to achieve the most efficient use of the channel over time. This provides the capability for maintaining a DSL connection during periods of difficult channel characteristics by lowering the throughput. This also enables the modem to increase the throughput and make the best use of the connection during periods of favorable channel characteristics. Ideally, the transceiver at each end can monitor the channel and maximize their throughput as conditions vary. A practical transmitter/receiver can be designed that increases or decreases throughput of the physical channel based on the available capacity, the available signal processing resources, and the requirements of the specific applications. Several rate adaptation methods exist (e.g. the standard CCITT V.34 Voiceband Modem Standard), but two particularly convenient techniques are

discussed later herein for two distinctly different modulation methods. However, the techniques for rate adaptation are easily extended to other modulation and coding schemes, and such extensions are considered part of the present invention.

Detailed Description Text (94):
Network accessibility in this context describes the rate and/or delay associated with the transfer of data from the local loop to the backbone network. This measure might be affected by the specific backbone network used (e.g. Internet, ATM, etc.), the bandwidth given by the service provider, and the amount of network traffic. The techniques defined in this invention are not restricted to use on a particular backbone network.

Detailed Description Text (95):
Although a VRDSL connection is capable of certain transmission throughput, the total throughput might not be connected to corresponding CO backbone networks at times. For VRDSL-provided services going through the PSTN (Public Switched Telephone Network), connections will be made only when services are initiated. For VRDSL-provided services terminated at the local CO, such as internet access, leased line or dial-up line connections with certain throughputs can be made depending on the preferred cost structure. The available CO backbone throughput to each VRDSL modem can be different at different times. The subscriber-desired throughput could also vary for different applications.

Detailed Description Text (96):
With actual throughputs lower than that provided by the VRDSL physical transmission Rink, traffic concentration can be realized at CO backbone networks. Statistical multiplexing can also be realized by using a separate analog front end for each CO VRDSL modem. The required number of corresponding digital portions can be less than the number of analog front-ends, depending on the traffic behavior. In the extreme case, the digital portion of the CO VRDSL modem can be multiplexed among active VRDSL links by using the voice-band as a traffic indicating channel and keeping a copy of the digital state portion of the modem inside RAM.

Detailed Description Text (97):
The VRDSL communications model is depicted in FIG. 7a. The sole purpose of this model is to aid in understanding the disclosed rate-negotiation technique. The model is composed of separate residence 7210 and central office 7220 layered representations of functional separation. The functionality of the residence terminal 7210 is shown on the left. The lowest layer 7330 is the Communication Hardware Layer, which contains the modulator/demodulator, signal conditioning, timing, synchronization, and error-correction coding. This layer can also be referred to as the data pump layer. The second layer 7320 is the Hardware Control Layer. This layer provides framing control and other data packaging functions that better organize the data received by the lower layer. The third layer 7310 is the Software Driver Layer. This layer provides an interface between the hardware levels and the application programs run at the residence. The fourth (top) layer 7300 is the Application Software Layer, which contains all functions provided by the application programs run at

the residence. This layer encompasses both the software to manage the throughput allocated to different simultaneous applications as well as the application programs themselves. Conventional software application programs request a channel and accept the available throughput provided by the lower layers (no negotiation). Future generations of software application programs might have the requirement and capability for rate negotiation.

Detailed Description Text (100):
A rate table is defined as a common syntax for the R and A signaling sign between layers. The rate table defines the rates that a particular layer can attempt to achieve. (In general, this will be defined by the hardware limitations of the modem.) During a rate request (R), an upper layer might signal a lower layer of a desire to change the rate structure. If the lower layer is able to reconfigure itself to a new set of operating parameters and achieve the requested rate, then it will do so and indicate this to the upper layer. If the lower layer determines the requested rate to be unacceptable, the upper layer is informed along with information about the rates that are available under the present operating conditions (A).

Detailed Description Text (106):
2. The Hardware Control Layers 7320, 7420 can view the communication link as a virtual 'Corrected' data stream. This is the actual throughput of the channel after the physical timing, synchronization, control, and error-correction coding redundancy symbols have been removed.

Detailed Description Text (107):
3. The Software Driver Layer 7310, 7410 views the connection as a virtual channel called the data link channel (DLC). For convenience, the DLC may be a frame structure that represents multiple N kbit/sec channels (N=16 or 64 e.g.). In addition, a control channel may be specified. This control channel may either be embedded in the lower layer channels or can be completely separated from the DSL connection. For example, the control signaling might be implemented hi the voice-band via a v.34 modem connection.

Detailed Description Text (109):
The basic requirement for rate adaptation is the rate table, a well-defined set of achievable rates that can be communicated to the upper layers of the DSL communication model. The rate-table is determined by the capabilities of the hardware at both ends of the connection. During startup or reset, a pair of modems must agree upon the rate table entries which they are both capable of supporting. The allowed rates under a given channel condition are then represented as legal states in the table. The different levels of the model can communicate via the rate-table syntax without concern for detail in other layers. This rate table can vary substantially from one modulation and/or coding scheme to the next, but the concept of allowed and disallowed rates depending on channel conditions does not change.

Detailed Description Text (111):
In the case of high-rate serial transmission of digital data,

digital symbols are chosen to represent a certain number of bits, say N. Groups of N bits are mapped into symbols which are transmitted over the channel. At the decoder, a decision is made to determine the transmitted symbols. If the correct decision is made, the transmitted bits are decoded correctly.

Detailed Description Text (112):
A method of changing the throughput changes the number of bits represented by each symbol while keeping the symbol rate constant. Increasing the number of bits represented in each symbol increases the number of transmitted bits, albeit at lower noise immunity. Decreasing the number of bits per symbol increases the noise immunity and improves the robustness of the transmission, but at the expense of a lower throughput. The bandwidth remains the same in either case.

Detailed Description Text (113):
Another straightforward method of varying the throughput is changing the bandwidth used in the transmission channel. By expanding the bandwidth, a greater number of symbols can be transmitted over the channel in a given interval. The symbol rate is roughly proportional to the bandwidth. However, the processing requirements of the DSL modem also increase with the bandwidth; higher bandwidth requires greater computation for modulation/demodulation. The maximum usable bandwidth might either be restricted by channel conditions or modem hardware processing capability constraints.

Detailed Description Text (115):
Let the nominal serial transmission rate be R. Define the minimum rate step by which a DSL modem can change as dR. If the minimum rate is R-2*dR and the maximum rate is R+2*dR, then the set of achievable rates is given by {R-2*dR,R-dR, R, R+dR,R+2*dR}. For example, let R=300 kilo-symbols/second, and dR=100 kilo-symbols per second. The set of achievable rates become {100, 200, 300, 400, 500} kilo-symbols/second.

Detailed Description Text (116):
Let N represent the number of bits conveyed by each transmitted digital symbol. For example, a VRDSL modem might support operation with N in the set {2,3,4,5}. The higher values of N will convey more bits in a given period, but will also result in lower tolerance to noise.

Detailed Description Text (119):
Discrete multi-tone (DMT) modulation transmits low-rate data symbols over parallel subchannels. By splitting a high-rate serial data stream into multiple low-rate data streams that are transmitted in separate subchannels, the system can be tailored to better match a frequency selective channel. Good portions of the overall bandwidth (those subbands with high signal-to-noise ratio (SNB)) are used to transmit symbols with a larger number of bits/symbol. An unequal number of bits are assigned to different subchannels, depending on the available capacity of each subchannel. Essentially, the data can be distributed among subchannels in a manner allowing very efficient use of the overall bandwidth.

Detailed Description Text (120):
As with the high-rate serial data stream, the overall bandwidth of a DMT system can be increased or decreased according to the overall desired throughput, channel conditions, and modem hardware capabilities. Additionally, DMT modulation provides the capability of dropping or adding bandwidth a single subchannel at a time. For a DMT system with a large number of subchannels, this creates a very large selection of possible bandwidths. If desired, the number of subchannels can be varied while keeping the overall bandwidth fixed.

Detailed Description Text (121):
For simplicity, consider a DMT system where the subchannel bandwidth remains constant, but the overall channel bandwidth used is controlled by the number of subchannels used. Let T represent the number of subchannels or tones used in transmission. Let N represent the average number of bits/symbol across the subchannels. N is no longer restricted to be an integer as with the high-rate serial transmission system. For this example, however, consider N to be approximately an integer valued. The following is an example of a rate table for DMT:

Detailed Description Text (122):
The parameter T represents the number of subchannels where each subchannel has a bandwidth of approximately 3.3 Khz. N represents the average number of bits/symbol represented fin all the subchannels. The table entries are given in kilobits/second.

Detailed Description Text (125):
The column parameters are labeled as different channel resource modes (cr1, cr2 . . . cr5), while the row parameters correspond to the average number of bits represented by each symbol. The entries represent the achievable rates for the 'Corrected' data stream in the VRDSL model.

Detailed Description Text (128):
A request for changing the current allocation of the data connection or channels in VRDSL such as requesting a new channel or changing an existing channel rate

Detailed Description Text (129):
When VRDSL physical layer detects a total channel capacity change either total channel capacity increase or decrease

Detailed Description Text (130):
After the initialization of VRDSL, a control channel (for example, of 16 Kbps) has been allocated as an initial channel connection. This control channel will be reserved during the whole physical line connection time. It is used to send/receive all the control information including rate negotiation information.

Detailed Description Text (134):
13 Channel map change Request

Detailed Description Text (135):
14 Channel map change Nak

Detailed Description Text (136):
15 Channel map change Reject

Detailed Description Text (137):
16 Channel map change Ack

Detailed Description Text (140):
Channel Map Data: The Channel Map Data Field is 2 or more octets which reflects the current channel allocation in the VRDSL line and the request for a channel change. It contains its own header and two parts of information represented by channel entry field:

Detailed Description Text (141):
Current channel map

Detailed Description Text (142):
Channel map change request

Detailed Description Text (143):
These two parts of information are all described by the 2 octet channel entry field. The way to distinguish them is that for channel map change request, the most significant bit of the channel entry is set high.

Detailed Description Text (144):
The Channel Map Data field is depicted in FIG. 7c.

Detailed Description Text (145):
When the code is 14 (Channel Map Change Nak), the Channel Map Data field contains: Total Capacity, Available Capacity, the current channel map and one or more channel entries which have been Naked. These Naked channel entries are flagged by their most significant bit (msb). When the code is 15 or 16, the Channel Map Data field contains: Total Capacity, Available Capacity and current channel map data.

Detailed Description Text (147):
The Link Layer Rate Negotiation is also called Channel Map Change (CMC) in VRDSL. A CMC procedure is described by the state change triggered by a specific event and action. FIGS. 7d and 7e depict state diagrams for the link layer rate negotiation during an active and passive CMC process, respectively.

Detailed Description Text (148):
Based on the VRDSL communication model, modem hardware capable of varying the transmission rate, and variable-rate management software, the rate negotiation method shown in FIG. 7f may be employed. FIG. 7f depicts a simplified functional diagram of the overall rate negotiation method.

Detailed Description Text (149):
Current QAM based voice-band modems make use of a handshake sequence between calling and answering modems to initialize their communications. To gain synchronization, the answering modem transmits alternating symbols of the corresponding constellation points. As an example, V.32 modems use the constellation points

A,B,C, and D in FIG. 8a in the synchronization process. The answering modem transmits alternating symbols ABABAB . . . for a duration of 256 symbols. After 256 symbols, the alternating symbols CDCDCD . . . is transmitted for 16 symbols. The transition period between the two symbol sequences provides a well-defined event that may be used for generating a time reference in the calling modem receiver. After the second symbol sequence the answering modem will start transmitting a symbol sequence that is known by both modems. This sequence is used to train the equalizer at the calling modem receiver. FIG. 8a depicts a V.32 training constellation.

Detailed Description Text (150):
The frequency response of the voice-band channel (30 Hz to 3.3 KHz) is nominally flat. The alternating ABAB . . . and CDCD . . . symbols can be reliably detected before equalization of the channel. However, this is not the case for the MDSL modem. For a 1/4 T1, modems use the spectrum up to 500 KHz of the telephone line. FIG. 8b shows the frequency response of a telephone CSA loop 6. A startup procedure that allows for partial equalization of the line is required before ting synchronization is attempted.

Detailed Description Text (151):
A preferred embodiment uses a startup handshake procedure for the MDSL modem. It uses an algorithm for implementation of the receiver portion.

Detailed Description Text (152):
FIG. 8c shows the time line for the proposed startup procedure for the CO and RU MDSL modems using CAP line code. The table below identifies the various segments of FIG. 8c.

Detailed Description Text (154):
CO MODEM

Detailed Description Text (155):
1. The CO modem is assumed to be always "on", but in an idle state. It continuously transmits segment A and listens for segment D.

Detailed Description Text (156):
RU MODEM

Detailed Description Text (157):
1. The RU modem comes on line and starts listening for segment A from the CO modem.

Detailed Description Text (159):
CO MODEM

Detailed Description Text (160):
2. Once the CO modem detects segment D from the RU modem, it transmits segments B,C, and valid data without further handshaking from the RU modem.

Detailed Description Text (161):
RU MODEM

Detailed Description Text (162):

3. The RU modem listens for segment B and once detected, it transmits segments E, F, and valid data without further handshaking from the CO modem.

Detailed Description Text (163):
4. The detection of segment B is the critical timing instant in the synchronization procedure. After it is detected, the RU modem begins training its equalizer using data from segment C.

Detailed Description Text (164):
CO MODEM

Detailed Description Text (165):
3. The CO modem listens for segment E from the RU modem. The detection of segment E is the critical timing instant in the synchronization procedure. After it is detected, the CO modem begins training its equalizer using data from segment F.

Detailed Description Text (166):
The receiver makes use of cyclical equalization techniques to obtain initial timing synchronization. On startup, the RU modem sets up a fractional spaced adaptive equalizer that is equal in time duration to K symbol periods, for example, K may be 15. This will be called the sync equalizer. If the sync equalizer is operated at two times the symbol period, the number of taps required is 2.times.K. For four samples per symbol period, the number of taps required is 4.times.K, and so on.

Detailed Description Text (169):
After rotation, the receiver continues to filter the signal, but does not update the sync equalizer coefficients. The output of the sync equalizer is then passed to a length K matched filter. The matched filter is used to detect segment B. Its coefficients are the transmitted channel sequence B. Since this sequence has only two values, a binary correlator could also be used.

Detailed Description Text (170):
When the output of the matched filter (correlator) is greater than a threshold. The receiver knows that the next symbol is the start of the training data. The receiver now implements the orthogonal adaptive filters used in CAP demodulation. They again are fractionally spaced adaptive equalizers whose lengths depend on the impulse response of the actual physical channel. These demodulation equalizers are trained using the known training data of segment C. After training has completed the demodulation equalizers enter a decision directed mode where the reference data comes from the CAP slicer.

Detailed Description Text (173):
For easy description purposes, the following notations are used: time domain equalizer taps w.sub.1 ; channel impulse response (including time domain do equalizer) h.sub.k ; the receiver data before the equalizer y.sub.m [n], and after the equalizer z.sub.m [n], where m denotes the label on data block. The received signals corresponding to the transmitted signals in FIG. 9a are as follows: frame number ##EQU4## where, p.sub.n is the pilot tone superimposing on the training sequence. The second terms on the

right hand side of the equations are attribute to the inter-symbol interference from the previous frame. The second term can be separated from the first term by performing the operation: frame 4.-frame 1. ##EQU5##

Detailed Description Text (174):
Assuming prefix length is L, the ideal channel impulse response is ##EQU6## The condition (2) can be satisfied if the time domain equalizer w.sub.1 is chosen such that

Detailed Description Text (177):
The frame boundary information can also be derived from above training sequence. As seen in Eq. (1), if the block of the training sequence is much longer than the channel impulse response, err[n] approaches zero as h.sub.N=k .fwdarw.0 when n increases to the end of frame 4. However, when data starts in frame 5, ##EQU10## For ADSL applications, since there is high attenuation in copper wire at high frequency, the channel impulse response h.sub.k does not expect to flip the sign very frequently. If the values of x.sub.n at the beginning of the training block {x.sub.n] have the same sign, the summation in equation 7 will be constructive. Consequently the amplitude of err[n] starts to increase at frame boundary n=0. FIG. 9b shows the time sequence of err[n]. As shown in FIG. 9b, the rising edge of the derived sequence err[n] can be used for frame synchronization, and the training edge of err[n] can be used for time domain equalizer training. For the same reason as that of in the rising edge of err[n], to make the summation in equation (1) constructive the elements of the training sequence at the end of block x.sub.N-k should also have the same sign.

Detailed Description Text (191):
The SLSD mode works on a switched MDSL line on which the MDSL-R modem is dialed automatically by the MDSL-C which is controlled by a remote server. Under this mode, the line management follows a special MDSL dial-up procedure that is independent from the Plain Old Telephone Service (POTS) line. The MDSL modem dial-up procedure is defined by the MDSL modem's internal initialization process. It has 2 dial-up IDs, one related to the MDSL-C port and the other related to the MDSL-R modem. The ID for MDSL-C port could be just the subscriber phone number plus 1 digit; by choosing it to be 0 and the ID for the MDSL-R modem could be the subscriber phone number also plus 1 digit selected to be 1. The other 8 values, from 2 to 9, are reserved.

Detailed Description Text (192):
The SLHD mode works in a way similar to that of voice-band modem but with MDSL dial-up procedure. The MDSL modem will either store a phone number or be dialed manually by an application.

Detailed Description Text (265):
After power on, the MDSL-R automatically precedes with its internal initialization process. This process contains four steps: channel probing, line code selection, rate negotiation and transceiver training. After the initialization procedure, the MDSL-R transitions to a stand-by mode. The line state at this moment is "disconnected" as defined before. Upon detecting that the line has been physically connected, the HOST software will send a

MdslLineConfigure() command to MDSL-R for line configuration. MDSL-R then sends out a line configuration command packet to MDSL-C with the configuration data. After receiving the line configuration command and checking the configuration data, MDSL-C will send out an acknowledgment packet to confirm the line configuration. If the MDSL-C cannot accept the configuration data, it will send a configuration reject packet. It will also give the status message specifying what kind error it is. If only part of the configuration data is not acceptable, the data field will contain the configuration data which is not acceptable, as depicted in FIG. 10g.

Detailed Description Text (334):
FIGS. 11a-b illustrate the software structure of the driver for modem 100 used with a host having a Windows 95 or Windows NT environment, as commonly would be the situation for a personal computer in a residence. FIG. 11c illustrates the software driver structure more generally.

Detailed Description Text (355):
The initialization entry point (MdslInitialize) will be called by NDIS library to initialize the MDSL modem.

Detailed Description Text (374):
2. Get ail the configuration information of MDSL NIC (interrupt number, board name, channel address or line address, switch type, etc.)

Detailed Description Text (577):
The software to configure a multimode modem as to its DSL band operation can be acquired by downloading into a Flash EPROM (see FIG. 5a of a board version of a DSL modem enhanced to include Flash EPROM). This downloading can be performed by using the voice-band configuration (V.34) already in the multimode modem. In particular, a host can use voice-band modem operation to call a source telephone number which then can download the software for DSL band operation over the voice-band to the Flash EPROM. In the same manner, updates of the DSL band software can be downloaded either over voice-band or over DSL band. FIG. 12 illustrates such a downloading process.

Detailed Description Text (578):
Referring now to FIG. 13a, there may be seen the MDSL frequency division for upstream and downstream. In voice-band modems, the highest frequency of interest is only 3.3 KHz. In MDSL, the highest frequency of interest can be hundreds of KHz. For example, for 1/4 T1 rates, the center frequency of the upstream channel F.sub.c1 is 100 KHz while the center frequency of the downstream channel F.sub.c2 is 300 KHz. The bandwidth of each channel is 200 KHz and the highest frequency of interest is F.sub.2+ =400 KHz. The challenge is to be able to process the data with a low cost programmable digital signal processor (DSP). This invention addresses how to reduce the processing requirements by making either passband signal depicted in FIG. 13a appear identical to the DSP.

Detailed Description Text (579):

The MDSL modem has two modes the central office (CO) and remote user (RU) modes. In the CO mode, the modem transmits in the upper frequency band and receives in the lower frequency band. In the RU mode the reverse occurs. The modem transmits in the lower frequency band and receives in the upper frequency band.

Detailed Description Text (580):
Using the normal interpretation of the Nyquist Sampling Theorem, a minimum sampling rate twice the highest frequency of interest is required to process the data. For the CO modem, the analog-to-digital converter (ADC) can sample the received signal at twice F.sub.1+. However, it must generate samples for the digital-to-analog converter (DAC) at twice F.sub.2+. For the RU modem, the DAC can run at twice F.sub.1+. However, the ADC must run at twice F.sub.2+.

Detailed Description Text (581):
This invention makes use of digital sampling rate conversion to decrease the sampling rate and consequently the processing requirements for the implementation of the MDSL modem.

Detailed Description Text (582):
For the RU modem, the high sampling rate is connected with the analog-to-digital conversion process. The 1/4 T1 example modem receiver front end is shown in FIG. 13b at the RU modem. The incoming analog signal, centered at 300 KHz is first bandpass filtered to maximize the signal to noise ratio by isolating the bandwidth of interest. The signal is then sampled by the ADC at the normal Nyquist rate of twice f.sub.2+, 800 KHz.

Detailed Description Text (585):
For the CO modem, the high output sampling rate is required in the digital-to-analog process. It would require a minimum sampling rate of 800 KHz to directly generate the output samples corresponding to the upper passband signal. It would be much better if the CO modem could generate the output samples in the lower frequency band, and somehow automatically translate the spectrum to the upper band. FIG. 13e shows the spectrum of the low band signal in the digital domain.

Detailed Description Text (586):
Translation can be accomplished by making use of the aliased images produced by digitally upsampling to a higher rate. Upsampling by two to 800 KHz consists of inserting a zero valued sample between the computed output samples. This generates images at harmonics of the original 400 KHz sampling Frequency. Even the new modified output data stream is passed to a DAC, the analog output spectrum shown in FIG. 13f is generated. (The since roll-off characteristic imparted by the conversion process has been left out of the figure). By the use of an appropriate analog bandpass filter, the inverted image centered at 300 KHz can be selected. Since the inserted values are zero, they need not be computed by the DSP. The inversion can be either corrected by multiplication of odd samples by (-1) or disregarded completely, since the spectrum is inverted again by the decimation process at the RU modem. As show in FIG. 13g, the zero sample interleaving process can be implemented by simple external logic outside the DSP.

Detailed Description Text (587):
In conclusion, the application of sampling rate conversion allows the DSP in the MDSL modem to assume that it is always transmitting and receiving only in the lower frequency band. Its computations are therefore based on a much lower sampling rate than would normally be dictated by the actual analog signal frequency content.

Detailed Description Text (593):
In the central office end, a modem pool can be used to handle multiple MDSL lines. Although a dedicated line coupling and front end circuit is necessary for each MDSL line, the signal processing power of a high performance DSP chip can be shared among multiple MDSL lines. The multiple line capability of an MDSL modem pool can be further enhanced by incorporating multiple DSP chips within a single modem pool unit.

Detailed Description Text (594):
FIG. 14a shows that an MDSL modem pool can have N logical MDSL modems, each consisting of a transmitter part and a receiver part. Due to the location of the modem pool, transmitters can be synchronized to the same central office clock. Because of the MDSL line concentration and the shared modem pool architecture, data symbols of the transmit signal and samples of the received signal are readily accessible among all logical modems. The transmit signal synchronization and the transmit and received signal accessibility enable the adaptation of NEXT cancellation technique. A multiple input-multiple output NEXT canceller can be implemented in conjunction with an MDSL modem pool.

Detailed Description Text (595):
To avoid the NEXT and the cost of echo cancellation hardware, a preferred MDSL modem uses frequency division duplex for transmission from a central office to a subscriber in the downstream direction and vice versa in the upstream direction. The downstream transmission normally occupies the higher frequency part of the MDSL spectrum. The frequency separation between the downstream and the upstream directions is based on the use of high order bandpass filters. FIG. 14b shows that a guardband is used between the upstream frequency band and the downstream frequency band spectrum. Furthermore, the bandwidth of each downstream spectrum can be different for different modems. This might be necessary because the spectral allocation could be optimized according to the individual line conditions and downstream to upstream throughput ratio.

Detailed Description Text (596):
Because of the finite amount of attenuation in the bandpass filter stopband and the closeness between downstream and upstream spectra, there will always be some residue noise from the reverse channel. Due to the heavy subscriber line attenuation, the relative strength of residual noise might not be negligible compared with that of the received signal. Because of the possibility of upstream and downstream spectra overlapping among different MDSL lines, the NEXT noise can occur within the region of guardband. Hence, the NEXT cancellation can be used to minimize the interference of reverse

channel residual noise of the same MDSL line and the interference of reverse channel NEXT noise from adjacent MDSL lines.

Detailed Description Text (597):
FIG. 14c shows that a reverse channel NEXT canceller bank can be implemented within the same MDSL modem pool unit with or without additional DSP chips. The NEXT canceller bank needs the access to the transmit signal and the digitized received signal of all modems. The NEXT canceller bank has N NEXT cancellers as depicted in FIG. 14d corresponding to N MDSL modems. Each canceller has N adaptive filters of size M. Outputs of all N adaptive filters are appropriately combined to form the NEXT cancellation signal for the corresponding modem. Each adaptive filter is adapted according to the error signal between the received signal and the NEXT cancellation signal and the corresponding transmit signal as the correlation vector as depicted in FIG. 14e.

Detailed Description Text (598):
Unshielded twisted pairs can be used for high data rate digital transmission. For the case of Digital Subscriber Lines (DSL) extensive digital signal processing, such as transmit signal shaping and received signal equalization, is employed to exploit the full capacity of the twisted pair transmission medium. Therefore, the cost of modems at both ends of the twisted pair become a significant part of the total cost of the transmission system. When the transmission distance is relatively short, a simple analog line driver is used for the transmitter and a simple threshold device is used for the receiver. By avoiding the use of extensive digital signal processing, the transceiver cost can be kept at a minimum level.

Detailed Description Text (599):
Using the observation that the high-rate high-precision A/D and subsequent high-precision digital signal processing is an expensive channel distortion compensation approach, a direct equalization method of the present preferred embodiment utilizes symmetrical twisted pair transmission channels that extends the transmission distance while keeping cost minimal. This direct equalization method can also be applied to non-twisted pair symmetrical transmission channels.

Detailed Description Text (600):
Many short distance twisted pair based transmission systems, such as 10 BaseT and 100 BaseT Ethernet, ATM 55 Mbps Physical Layer, IEEE 1394, IEEE 1355, etc., have a symmetrical channel response. Because there are no bridged taps, channel transfer functions in opposite directions are identical for time-division duplex systems. In these cases, the channel response can be identified by examining the received signal. Specifically, it can be identified by transmitting training sequences during idle time between data transmission periods. Easily distinguished binary training sequences are preferred for channels with severe distortion.

Detailed Description Text (601):
FIG. 15a shows the channel transfer function of a 50 meter 24 gauge twisted pair. The channel distortion is caused by differences in attenuation and phase delay at different frequencies. The channel

distortion causes intersymbol interference which cause the eye pattern to close. FIG. 15b shows the eye pattern of the received signal at the end of a 50 meter 24 gauge twisted pair. The degree of the eye pattern closing can be judged by the relative signal level spread at the maximum eye opening point. The time interval at which the transmitted signal levels can be reliably determined is also very important. In a practical system, timing jitter exists between the transmitter and the receiver. A wider available decision window would decrease jitter requirements.

Detailed Description Text (602):
The eye pattern closing caused by channel distortion can be compensated by the use of a channel equizer. Specifically, the distortion compensation at the baud rate will reduce the spread of signal level at the optimal decision point. Furthermore, the above baud rate distortion can expand the available decision window. In other words, a baud rate equalizer can only maximize the eye opening at a particular decision point while a fractional spaced equalizer can maximize the eye opening at more than one point, thus expanding the optimal decision window.

Detailed Description Text (603):
A traditional channel equalizer implemented in the receiving path of a transceiver is seen in FIG. 15c. The received signal is amplified 1510 and converted into digital format 1512. A programmable filter 1514 with adjustable coefficients 1516 is used to compensate the channel distortion. These filter coefficients 1516 are calculated to minimize the mean squared error between the filter output signal level and the desired signal level. The calculation can be carried out based on the Least Mean Square (LMS) algorithm. Slicer 1524 quantizes to signal levels for decoding. Output data is converted to analog 1536, and line driver 1538 transmits them with switch 1520 providing isolation.

Detailed Description Text (604):
The realization of the conventional equalizer usually requires a full-precision programmable filter. Depending on the channel distortion and the number of signal levels, an A/D converter 1512 with 6 to 10 bits of resolution is necessary. This A/D converter must operate at or above the symbol rate. A baud rate based channel equalizer ranging from 10 MHz to 30 MHz also needs a highly accurate timing recovery circuit. The programmable filter 1514 after the A/D converter should have the same or higher bit resolution in the data path to make the equalization process effective. The high resolution and high operating rate A/D converter and the following programmable filter of the same resolution and the same operating rate translate into a high transceiver cost.

Detailed Description Text (605):
In a noiseless environment, the equalization function that compensates for the frequency distortion of the channel can also be performed using a programmable filter in the transmitter. These transmitter filter coefficients are adapted in real time by using a training sequence. During data idle time periods, a bi-level training sequence is transmitted for the purpose of filter coefficient adaptation. The receiver correlates the received

training sequence with the known training sequence and updates the equalizer filter coefficients using an adaptation algorithm, such as the Least Mean Squared (LMS) algorithm. Since the channel is symmetrical, the identified equalizer coefficients are then used for the programmable transmitter filter.

Detailed Description Text (606):
The direct equalizer system of the present preferred embodiment includes a transmission path and a receiving path, as depicted in FIG. 15d. In the transmission path there is a switch 1534 controlled by the data buffer status to multiplex the training sequence 1540 and the data. When the data buffer is idle the training sequence is linked to the D/A conversion device 1536. To avoid transmission collision, a higher layer protocol algorithm is also necessary to regulate the transmission of training sequences at both ends. In the receiving path, a received data detection function is necessary to control the adaptation of the transmitter filter coefficients. The combination of the transmit filter 1532 and its adaptation mechanism forms a direct channel equalizer. The filter coefficients can he updated periodically using a Digital Signal Processor (DSP) in a few baud intervals.

Detailed Description Text (609):
A baud rate equalizer, either at the receiver or at the transmitter, can only compensate for channel distortion at its precise sampling points. Hence, a receiver needs an accurate timing recovery circuit to keep track of these optimal sampling points (FIG. 15f).

Detailed Description Text (610):
The optimal sampling window size can be expanded using a fractionally spaced direct channel equalizer. FIGS. 15g and 15h show the effects of fractionally spaced direct equalizer for equalizer operating rates of 2 times and 3 times the baud rate. Above baud rate direct channel equalizers can create optimal sampling window sizes of 50%, 66.6%, or more.

Detailed Description Text (617):
The direct equalization approach will enhance the spectral density at the high frequency portion of the transmit power spectrum. However, depending upon the channel characteristics, the enhancement should be in the range of only a few dB.

Detailed Description Text (618):
The power spectrum of the multiple level Pulse Amplitude Modulation (PAM) signal is ##EQU15## where A is related to the signal amplitude and f.sub.0 is the baud rate. The frequency response of the equalizer is ##EQU16## where H(f) is the channel transfer function and T.sub.s is the equalizer operating sample interval. One has ##EQU17##

Detailed Description Text (619):
For a typical twisted pair channel, one has

Detailed Description Text (622):
FIG. 15j shows a simulation system for the direct equalizer using PAM signals. In particular, the transmitted PAM signal is delayed

(z.sup.-k) and fed to receivers along with the through channel received signal. These receiver1 signals are used to control the transmitter for PAM signal2 to directly equalize; and this compensated signal through the channel at receiver2 is compared to the PAM signal2. FIG. 15k shows the interior of receiver1 of FIG. 15hj, and FIG. 15l shows the interior of the transmitter of FIG. 15j.

Detailed Description Paragraph Table (1):
TABLE 1

Channel Capacity vs. Modulation Type Theoretical Practical # Bandwidth Practical Estimated #3 Chs. with Modulation Efficiency Bandwidth Mbps System Sectorized # Type FEC Encoding (b/Hz) Efficiency Channels Factors of Chs.

QPSK R1/2 1 0.8 192 192 768 QPSK R2/3 1.3 1 240 240 960 16QAM R3/4 3 2.4 575 288 1152 64QAM R5/6 5 4 960 480 1920 16VSB R7/8 7 6 1440 720 2880

Detailed Description Paragraph Table (5):
Segment Description
A, D One orthogonal channel is a repeating K-symbol sequence using the maximum value of the CAP constellation. For 16 constellation points, the channel can take on the values of +/- 3. The other orthogonal channel is a random sequence using all possible points of the CAP constellation. For 16 constellation points, the channel can take on the values of +/-1, or +/-3. B, E One orthogonal channel is a length K sequence that is the inverted version of the K-symbol sequence used in segment A. The other orthogonal channel is a length K random sequence using all possible points of the CAP constellation. For 16 constellation points, the channel can take on the values of +/- 1, or +/-3. C, F One orthogonal channel is a length L random sequence using all possible points of the CAP constellation. For 16 constellation points, the channel can take on the values of +/- 1, or +/-3. The other orthogonal channel is a length L random sequence using all possible points of the CAP constellation. For 16 constellation points, the channel can take on the values of +/- 1, or +/-3.

CLAIMS:

1. A data rate negotiation process for a pair of modems which comprises the steps of:

(a) providing a subscriber station having a first modem;

(b) providing a central station having a second modem;

(c) transmitting a signal from said first modem to said second modem requesting a first data transmission rate;

(d) transmitting a signal from said second modem to said first modem either accepting or rejecting said first data transmission

rate;

(e) responsive to rejection of the requested first data transmission rate, repeatedly repeating steps (c) and (d) with a continually different requested transmission rate until said second modem accepts a said requested data transmission rate; and

(f) then commencing non-rate negotiation communication between said first and second modems at the accepted negotiated rate; wherein rate adjustment is based upon a data rate table associated with each of said first and second modems indicating available data rates for respective modem, and further based upon at least one of line conditions between said first and second modems, backbone network accessibility, computational capabilities of the modems, and application requirements.